

THE SOUND ENGINEERING MAGAZINE JULY 1969 75c

Television Sound: Panel Discussion The TV Audio Mixer West Coast AES Picture Gallery



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# Coming

• John Borwick, noted British writerhas contributed the first of what will be a series of European newsletter-type articles. It's title is DOLBY REVISITED.

Robert C. Ehle has a new article on electronic music systems called a REAL TIME COMPUTER SYNTHESIZER SYSTEM.

A COMMON BASS MIXER—FILTER AMPLIFIER. is the title of a circuit article by Walter Jung.

And there will be our regular columnists, George Alexandrovich, Norman H. Crowhurst, Arnold Schwartz, and Martin Dickstein, coming in **db**, The Sound Engineering Magazine.

> About the Cover



• Dramatic lighting enhances the clean lines of this console photographed at the Audio Designs booth at the West Coast AES Convention. In the photo above, Audio Designs president, Robert Bloom (closer to the camera) is explaining the features of the console to noted recording engineer William Robinson (behind). The AES Picture Gallery begins on page 28. <section-header><text><text><text>

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George Alexandrovich Sherman Fairchild Norman Anderson Prof. Latif Jiji Daniel R. von Recklinghausen William L. Robinson Paul Weathers John H. McConnell

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# Letters

The Editor:

I have been following your articles on the quality of TV audio with great interest.

Mr. Canby implied in his article in January that the audio portion of the TV broadcast is not always what it should be. Here is the only point on which I can agree with him, but not for the same reasons. He states that "the audio signal in television should be rigidly perfect, or as near to perfection as the art allows." But he then reverses his position and gives a multitude of reasons why it is almost advantageous, nay mandatory to have low quality TV audio! Mr. Canby certainly must know that there are certain FCC regulations governing the audio portion of the TV broadcast that must be adhered to. For instance, minimum requirements for the sound portion of the TV transmitter are a bandwidth of 50 to 15,000 hertz with a harmonic distortion of not more than 3.5% and a signal-to-noise ratio of at least 55 dB. This would certainly seem to conform to Mr. Canby's original statement. Therefore, any poor audio quality must come from a prior source. Television audio comes from three main sources, optical tracks on film, magnetic sound (either film or tape), and finally "live' sound from the studios. Let us start with optical sound for two reasons; first, over half of the audio on TV comes from film, and second, Mr. Canby intimates that it must be of high quality to match "the sharpness and definition" of the movie screen. Yet if Mr. Canby had done his homework he would have found that optical soundtracks have limited bandwidth (8000 Hz on 35 mm. and 6000 Hz on 16 mm.), suffer from relatively high cross-modulation and other distortion, and have poor transient response. In addition, it is common practice to compress and limit the average track so that everything comes out at the same level. So if optical sound is transmitted, it certainly is not deteriorated by the TV transmission system but is rather marginal to begin with. Far superior sound comes from magnetic film or tape. And of course, the best sound comes from the "live" TV studios. The care and trouble taken by the audio engineers to produce a high-quality sound is beyond all criticism. From microphone to mixing console to high-quality audio monitors, every effort is made to produce the best quality sound possible.

But even this high-quality sound is for nought when received by the average monochrome or color receiver. For now we come to the real culprit, the TV set manufacturer, who in the interest of economy has cheapened the audio portion of the receiver to the point where the end result is barely passable. One may ask, why doesn't the public complain about this poor audio? The answer is simply that the puble has been fed so much poor quality on AM radio that the sound of the average TV set is far superior to the pockettransistor sound that the listener accepts everyday. An interesting parallel exists in the elimination of the d.c. restorer in that same monochrome set. Here again the set manufacturer is to blame, he is again saving perhaps a dollar a set by eliminating d.c. restoration in the picture. And yet, the lack of proper grav-scale rendition in the picture is neither missed nor lamented by the general viewer.

Now instead of giving aid and comfort to the TV manufacturer by agreeing that no improvement in audio is needed nor desired, Mr. Canby should be in the forefront of those audio purists who demand only the highest-quality sound with their pictures. But no, Mr. Canby introduces his "audio-visual mix" to prove what we need is worse audio to match a "tiny, fuzzy little TV picture." If Mr. Canby is used to seeing this kind of picture, I can only suggest fixing his old set or getting a new one. For it has been proven in literature that our TV system is capable of transmitting excellent high-quality pictures. Schade of RCA proved long ago that our 525-line picture can be the equal of 35 mm. motion picture film in terms of picture sharpness.- Mr. Canby confuses sharpness with definition, although there is very little relation between the two. He admits on one hand that "it is obviously an adequate standard for a vast range of entertainment and useful message transmission." But he then states that "TV is flatly lo-fi" (without defining his terms) and this is absolutely untrue. It has been proven that the *fidelity* of the TV system in such objective terms as aperture response, light-transfer characteristics, signal-to-noise ratio, etc., is the equal of 35 mm. film.- Mr. Canby continues that since TV is "lo-fi" it deserves audio with less density than the picture. He mentions the audio-visual mix" as the reason for this which he says has to do with compatibility (or similar attributes) of a dual-sense message. I would appreciate learning where this Robert Bach PUBLISHER

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db July 1969

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theory arose and see the proof for it. Mr. Canby states, "that as audio quality improves, there is an increasing discontinuity between audio and  $\tau v$  signals." This simply is not so, there may be a dominant signal, but it is *always* compatible with the other senses. The total viewing situation makes specific demands on each sense. Therefore, the worsening of the audio component, for example, simply deteriorates the entire viewing situation, as conversely a high-quality audio component will enhance the total viewing situation.

Proof of this exists in the movie house where the average quality sound is enhanced by the large screen picture. Yet Mr. Canby states that, "the 'oldie' sound films now prevelant on TV in re-run form makes a better soundpicture mix than these same films ever did in the theater." Yet these pictures have the same quality sound track as they had when originally shown. Since the aural and visual senses are always compatible, one would not be aware of the rather poor sound on these films unless they exceeded certain levels of tolerance. However, Mr. Canby states that no one is aware of the sound deficiency because, "the TV picture reproduction is nicely deteriorated for an ideal blend". This is nonsense since it is known that the TV system is capable of reproducing both picture and sound quite accurately. A "nicely deteriorated" picture would be only too apparent and would certainly detract from the total viewing situation. Additional proof exists in the TV studio. If Mr. Canby were to come to a live (or tape) telecast he would see an amazing phenomenon. With pure. live audio whether speech, orchestral, or choral, spread over a wide stage he would be amazed at the percentage of people who insist on watching the monitors overhead rather than the live performance on the stage. Here we obviously have a "mix" of TV pictures with the best, no distortion, stereophonic, directional sound existing: live audio. Yet, there is none of the confusion Mr. Canby states. the "fighting to reconcile two utterly unrelated images of the same thing"! Obviously this destroys his theory of the "audio-visual mix" for the live audio enhances the total viewing situation. Another example is to be found in the viewing or editing room of this same studio. Only now the audio comes from high-quality speaker systems, (monophonic to be sure) and again the total effect is quite pleasing, no confusion, no wall-eyes or cross-ears, although we do get the same question as to why the program can't sound that good at home.

Mr. Canby's remarks regarding stereophonic sound on tv can thus be disregarded for the reasons given above. Perhaps the introduction of a multiplexed TV sound system will give the set manufacturers an excuse to introduce high-quality sound (at a higher price of course). For stereo sound is but one more step in the evolution of television that will ultimately include stereo sound as well as stereoscopic pictures. A recent broadcast magazine states that simulcasts of the Detroit Symphony Orchestra in full color with the audio transmitted by means of FM stereo have already been transmitted in the Boston area by the combined facilities of WGBH and WGBH-FM.<sup>3</sup>

#### **REFERENCES:**

- Schade, O. H. Electro-Optical Characteristics of Television Systems, RCA Review, vol. 9, no. 4, p. 653, Dec. 1948.
- Abramson, A. Picture Quality: Film vs. Television, jour. SMPTE, vol. 77, no. 6, pp 613-621, June 1968.
- Maynard, H. FM: Free Music, Faithful Market, Broadcast Management/Engineering, vol. 5, no. 2, p. 32, Feb. 1969.

Albert Abramson Van Nuys, California

#### Mr. Canby responds:

Yes, I admit to generalizations. To generalize is often useful and sometimes a necessity, a matter of seeing the forest in spite of the trees. Whether my generalizations are false or not is another matter. Mr. Abramson seems to me to have wholly missed, or dodged, my point in several instances. It is a well known super-fact that facts may be used to prove almost anything one wishes (including, as I remember, that a bumble bee can't fly). I would not presume to challenge Mr. Abramson's array of actual evidence, though it is untrue that I was not aware of a good deal of it. After many years in and around the business, I am quite aware of the varying quality of sound sources, for example.

First, in respect to TV audio transmission, I did not "reverse my position". I feel that, as in disc records, the purely professional segment of the TV transmission should be letter-perfect and up to accepted standards, quite aside from either the source or the ultimate home reproduction. That was my point. The analogy with discs is exact, and pertinent: there, too, the transmission from source to pressed record should be of professional quality, regardless of source *or* possible re-producing equipment. As every audio man knows, this has not always been the case, often by deliberate choice of the manufacturer, who reasons either that the public "doesn't like" good sound or that the home equipment, being faulty, must be catered to. To



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an extent, yes (equalization, for example); but not at the expense of good reproduction on good equipment. All of us have been through this set of arguments, nor do I need to quote voluminous facts (and manufacturers' names).

I did not say it was "mandatory" to have low-quality ty audio! Far from it! I merely point out some of the subjective effects of our present audiovideo "mix" as it actually appears on home screens and via home speakers. Subjective reactions are not easy to prove-which does not make them any less important. Not all will agree with me that the effective clarity of sound and picture must be approximately alike for the best combined transmission of the message. Some most surely will agree with me. I do not think the idea should be by-passed; it is surely relevant and very possibly there are those with more professional knowledge than myself who can supply factual evidence of my contention. My hope was to offer constructive thought on what is at best a poorly understood area of communications.

One important additional factor has been suggested to me, which could well further confuse the issue. We are not all hearing the same TV audio on a given program. Those of us who live near the major TV centers normally receive direct broadcasts. But many viewers in the less populated parts of the country get their entertainment via such highly restrictive links in the audio chain as limited class telephone lines. This, of course, over and above the variations in source and, at the other end, the quality of the reproducing equipment.

Yes, it has been proven in the literature that our present video picture standard "can be the equal of 35mm. film in terms of picture sharpness." Perhaps in all truth it is "obviously an adequate standard. . ." Strange, for one thing, that the Europeans have adopted a more exacting standard. Why? Strange, too, that almost any viewing eye can see that the average video picture as received in the home is not equal to a 35 mm. film projection! Nor is the distinction between sharpness and definition a fundamental one in this respect. I leave this particular argument to those who can best carry it forward; I only suggest that it is unwise to assume that present video is ideal in practice, as well as in theory.

I am afraid I must stick to my subjective point that "as audio quality improves, there is an increasing discontinuity between audio and TV signals." For my ear and eye, it is true, and there are those who will back me up. As for the dominant signal and the necessity for greater over-all clarity (I use that word deliberately) to exist in its message, the observation is my own and I can give no antecedent source though I suspect you will find similar ideas expressed in the MacLuhan writings. I have some faith in my own senses, if others do not, in theirs.

The large-screen movie image in the theatre carries the inadequate sound of the older movies along with it-that is according to my idea. But I still maintain that a better practical mix is realized when the picture is seen via TV. To put it another way, one is aware of the sound's inferior quality when the movies are now shown in the theatre; on TV one is less likely to notice any lack, in a conscious fashion. That is, of course, all to the good as far as the message is concerned. What is to be avoided at all costs is any sort of conscious discrepancy, or irrelevancy, between sound and sight. The "audiovisual mix" most surely does exist, whatever Mr. Abramson may say, and the fact that it is by nature a highly complex and subjective phenomenon does not give us the privilege of ignoring it.

As for stereo, and Mr. Abramson's account of studio and monitor viewing, I feel that his points are essentially irrelevant. Studio-audience TV is a very special thing, as is professional monitoring; what is important is the "proof of the pudding", the television we re-ceive at home. I will leave my arguments as to stereo to stand on their own merits, and hope that perhaps others will take over for me in the discussion as to whether TV stereo can be effective-and how. I would be delighted if a way could be found to combine stereo sound with pictures-especially, stereo pictures of some sort. But the combination involves extremely complex and subtle inter-relationships. In personal experiments over many years I have found that although direct (i.e. lip-synchronized on-the-spot) binaural recording makes a superb sound-picture blend, regular two-channel stereo sound is simply mono with added complications (as I have described them) which perhaps more than balances the advantages, in practical TV terms. I am sure we would all welcome constructive argument on these points since  $\tau v$  stereo, and even "3-D"  $\tau v$  pictures may actually prove to be both practical and, hopefully commercial.

In sum, it seems to me dangerous to assume anything at all as "proved" in this day of rapid communications development. Ideas—even such as mine which can throw light and, perhaps, raise useful doubts, may in the end serve their purpose even though proved, if not false, at least impractical. Mr. Abramson's contribution in this respect is surely as useful as mine, and I thank him for his attention.



# The Professionals.

Bill Bell is known as "The Ear." He's the owner of Bell Sound Studios, Hollywood. Bill does commercials, some of the best. You've heard a lot of them. He orchestrates each one, every element of sound from the soft spoken solo voice of Marvin Miller to the high dB blare of acid rock.

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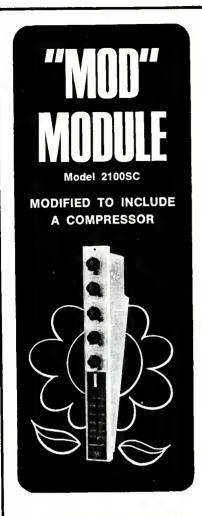
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# The Audio Engineer's Handbook

## GEORGE ALEXANDROVICH

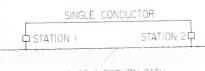
### BALANCED AND UNBALANCED LINES

•How many times have you asked yourself, "Should I use balanced or unbalanced lines?" Before we get into any further discussions about the use of the lines, let us look at balanced and unbalanced lines in general.

Transfer of electrical signals from one location to another can be accomplished only by means of a closed loop, meaning that for energy being sent out, an equal amount of energy with the reverse polarity should be received back. If we talk about electron flow, the amount of electrons being sent into the line should be replenished through the incoming line, thereby producing the current flow. We talk here about two wires; one, as being used to send signal out, and the other which acts as a return path for the same signal.

Signal currents in the audio-frequency range changes its direction many times per second, but the direction of current flow in two conductors is always opposite to each other.

All audio transmission lines work on the same principle. In the case of field telephones for instance, one wire is omitted and ground is used as a return path (FIGURE 1). Therefore, basically only one wire (hot) has to be strung. This single conductor acts pretty much as an antenna, picking up random electrical fields surrounding it and mixing them with the original signal being sent through the line. The ability of



GND AS A RETURN PATH

Figure 1. An unbalanced line such as is used by field telephones.

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this transmission line to pick up extraneous harmful fields causes electrical interferences. Since one side of such a transmission line is at ground potential and the other is not, we call such lines unbalanced (FIGURE 2). Unbalanced it is, with respect to ground and external fields. One way to protect such a line from picking up external interferences is to isolate it from external fields. This can be accomplished through shielding. In order to minimize the effect of external fields, insulated transmission wires are kept close together with flexible metal shield in the form of a tube surrounding them. This shield is normally grounded at either one or many points.

If this shield were near 100 per cent effective, it would have to be made of high-quality magnetic shielding material with an ability to block all electrostatic and magnetic fields as well. Shields made out of aluminum, copper, or other non-magnetic (nonferrous) materials stop *almost* all electrical interferences caused by electrostatic and radio-frequency fields but don't protect the wires from picking up low-frequency magnetic fields from a.c. transformers, power lines, motors, solenoids, and other sources).

Since there are no practical shields which protect the unbalanced transmission lines completely for all types of interferences (and in order to accom-



Figure 2. The unbalanced interconnection of amplifier stages.

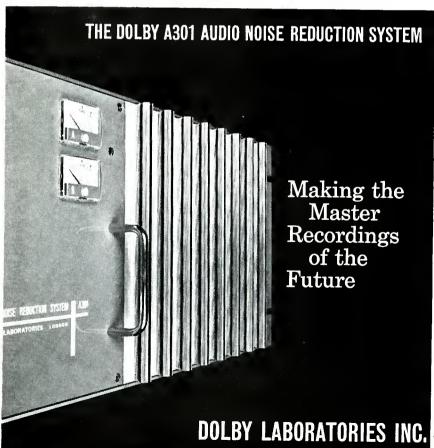
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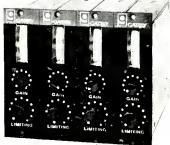
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plish transmission of low-level signals economically over long lines) a method of balancing transmission lines at both ends is being used. This method of balancing has been known for decades, the largest user of balanced lines are telephone companies. Almost all telephone installations and exchanges use balanced lines which allow use of unshielded wires with relatively low losses (FIGURE. 3).

In order to balance the line, transformers must be used at each end of the line. Transformer windings connected to the balanced line are made with the center tap normally grounded, at least at one end. The idea behind this hookup is to keep both wires of the balanced line at the same potential with respect to the ground. External fields affecting these lines, obviously affect both sides of the line equally at the same time, although both sides of the line may be saturated with external fields. Signals arriving at the terminating transformer cancel each other out.

The effectiveness of a balanced line depends on two factors. *First*, on the transformers and their symmetry of windings; *second*, on the transmission lines and the symmetry of the interference signals induced in both sides of the line.

Before we start saying when we should use balanced and when unbalanced lines, we ought to examine the advantages and disadvantages of each.

Balanced lines are obviously less sensitive to the external interference but require transformers which are costly, bulky, and can pick up external fields themselves. Unbalanced lines are more economical to use over shorter distances but require much more care in connecting two pieces of equipment in order not to cause any harmful ground loops. Unbalanced lines also are more susceptible to external interference. Balanced lines are cheaper for longer transmission hauls because they do not require shielding. Their interwire capacitance as well as capacitance to ground can be kept small (if space is not premium) but they require good transformers which impose limitations of their own on frequency response, level, noise, distortion, and phase.

Unbalanced lines are very costly on long hauls but very economical when used within the system. They are very convenient when used with transformer-less transistorized equipment and do not present limitations on electrical performance found in balanced circuits. A disadvantage (but many may think of it as an advantage) is the ability of a balanced line to have its phase inverted 180 degrees, something that is harder to do with unbalanced lines unless a transformer is used or a phase inverting amplifier.

We come to the point where we



# Figure 3. The use of a transformer for the conversion of a line to and from unbalanced to balanced.

should decide on an arbitrary set of rules guided by the state of technology in the field of audio, regarding the time and place for the use of balanced and unbalanced lines. (Let us just add before any final deductions are made that in both cases the amount of interference picked up by the line is proportional to the length of it. Other factors may have an influence on the amount of pickup, such as the curvature of the lines which may have some cancelling effect, or ground capacitance which would have tendency to attenuate any rf type of interference. These factors will be neglected at present.)

The set of rules we have to establish are to be developed based on the experimentation and experience of many audio specialists and engineers in this field. Let us start with microphone lines.

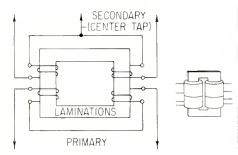
Don't ever attempt to use unbalanced mic-input lines unless they are shorter than a few inches. This includes the mic for talkback system within a console, announcer mics on goosenecks with the mic wires going directly into the preamp input, or any microphone with built-in amplifier (fet condenser microphones which may have output levels far exceeding levels of conventional microphones). Although it is possible to run some mic lines unbalanced, no self-respecting professional would dare to run these lines any other way than balanced.

In the cases where inter-connection of several pieces of equipment with unknown ground potentials and different power supplies is expected, balanced lines are mandatory. This refers to patch bays in particular. You never know what equipment you will be called upon to patch in and the only sure way to prevent ground loops, noises, and melted patch bays, amplifiers, and power supplies is to use balanced lines isolated with transformers.

The trends of today's designs are to process the signal from the source to the final destination with the least amount of deterioration of quality. In this age of economical awareness, miniaturization, and high performance standards, the transformer is the most



Figure 4. A shielded unbalanced line using a double conductor within a shielded shell.



#### Figure 5. A balanced transformer.

objectionable part of the system, so a tendency exists to eliminate it from as many circuits as possible. I think this trend against the *indiscriminate* use of transformers is a healthy one, but I also think that we will never get to the point where transformers can be entirely eliminated. Let us consider recording or broadcast consoles.

The only place experience dictates us we should use transformers for isolation and balanced lines is on all inputs and outputs from the console. (Except for the direct monitor speaker lines not going through the patch bays). All the rest of the circuits within the console can be unbalanced and there is no other reason than having access to parts of the circuitry balanced, to do otherwise (FIGURE 4).

I doubt very much that you will find any patch bays used for access to the individual components within the system in recently designed consoles. Most of the circuits use switches rather than patch bays, eliminating the need for transformers.

It is sad to note that there has been no successful attempt made over the vears to improve transformers except for some minor improvements in alloys for laminations. While other components such as capacitors, resistors, switches and transducers have undergone radical changes in their designs and performance, transformers from today are little better than from the 1930's. There are no balanced mic transformers of quality readily available off the shelf which do not have to rely on triple magnetic shields to eliminate hum pickup, rather than having their both halves of primary and secondary windings physically positioned out-of-phase so that all external fields induced into the transformer would cancel out (FIGURE 5).

Foreign countries use such transformers extensively. We don't. Nor do we have any transformers which have their coils completely surrounded with laminations acting as a magnetic shield? Nor do we have laminations, without the gap (except for expensive toroids). It looks like we have to do some catching up with the rest of the world. Perhaps we should learn to be more perceptive to better ideas, designs, and practices.



An impossible dream?

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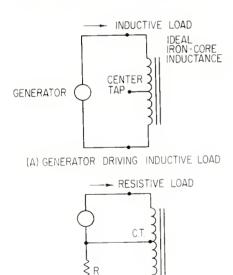
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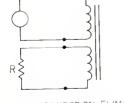
# The Feedback Loop

# ARNOLD SCHWARTZ

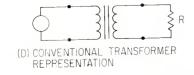
• Of all the questions that I have been asked about electrical circuits, there is one that has occurred more often than any other. That question, or rather category of questions, has been about audio transformers. The following discussion of the important features of audio transformers is in response to all of these questions. Before writing this article I took the opportunity of spending some time at United Transformer Company (UTC) to see how one of the



(B) GENERATOR DRIVING RESISTIVE LOAD



(C) ELECTRICAL CONNECTION ELIMINATED



db July 1969

2

leading audio transformer manufacturers designs and tests their audio transformers.

An interesting approach to transformer operation is to start with a generator driving an ideal iron-core inductance. See FIGURE 1(A). The generator will see an inductive load. By connecting the generator from the center tap to one end, and a resistor from the center tap to the other end, as shown in FIGURE 1(B), we have converted the inductance to a transformer. The phase of the current in each half of the inductance is such that the inductive reactance is cancelled out, and the generator will see a purely resistive load. This will only happen with a coupling coefficient of unity. Later on we shall see the effect of less than unity coupling. The electrical connection at the center tap need not be made as shown in FIGURE 1(C), leaving only the magnetic coupling between the two windings. FIGURE 1(D) shows the transformer in its more conventional representation.

### IMPEDANCE, VOLTAGE, AND CURRENT TRANSFORMATION

Transformers in audio signal circuits

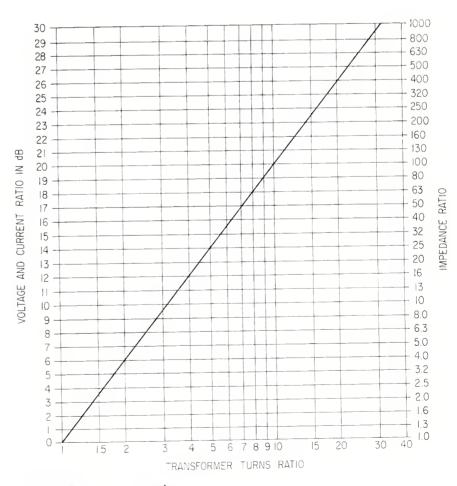
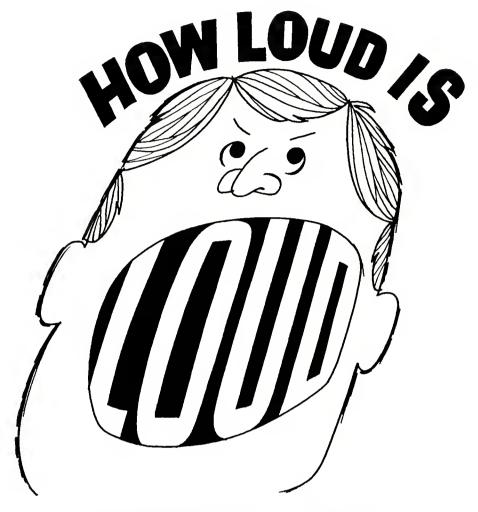


Figure 2. Transformer nomograph.

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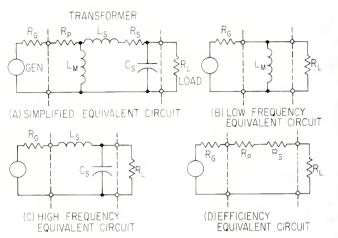






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Figure 3. (A) A simplified equivalent transformer circuit; (B) is a low-frequency equivalent circuit; (C) a highfrequency equivalent circuit; (C) a high-frequency equivalent circuit; and (D) an efficiency equivalent circuit.



are used for isolation, impedance matching, voltage transformation, and current transformation. Voltage and current changes are a function of the turns ratio, and the power in the primary circuit equals the power in the secondary. We therefore have the well-known transformer equations:

$$\frac{N_{p}}{N_{s}} = \frac{E_{p}}{E_{s}} = \frac{I_{s}}{I_{p}} = \sqrt{\frac{Z_{p}}{Z_{s}}}$$

$$N = Turns \qquad I = Current$$

$$E = Voltage \qquad Z = Impedance$$
subscript p = primary

subscript  $_{s}$  = secondary

These equations are set down in the form of a nomograph (FIGURE 2) which can be used more easily to determine the ratios of primary to secondary turns, impedance, voltage, and current. The current and voltage ratios are expressed in dB. Very often a transformer or circuit specification will only list one of these items, but with the nomograph of FIGURE 2 we can determine the remaining primary and secondary relationships. Take the case of a transformer used to match a 150-ohm line (primary) to a 600-ohm line. The impedance ratio is 4:1. We enter the nomograph at the right hand side, where the impedance ratios are shown, at the indicated ratio of 4. Going to the left on that horizontal line, we intersect the voltage ratio scale at 6 dB — which is the voltage gain and also the current loss in the secondary. To find the turns ratio we note where the horizontal 4:1 impedance line intersects the sloping line; we then drop down vertically from that point to the turns-ratio scale at the bottom of the nomograph. We intersect this scale at 2, so that we know the turns ratio is 2:1. The secondary has the larger number of turns since it is a step-up transformer. As a second example, we can start with a known primary to secondary voltage gain of 10 dB. We enter the nomograph on the left hand side at 10 dB, and going to the right on that horizontal

line we intersect the impedance ratio scale at 10 which means that the secondary impedance is 10 times that of the primary. The turns ratio is found as in the first example and is slightly less than 3.2:1 (actually 3.17:1). The nomograph can also be used to convert voltage or current ratios into the dB equivalent.

### TRANSFORMER PERFORMANCE

Up to this point, we have assumed a perfect transformer. Unfortunately, as in all other things, the perfect transformer does not exist. We therefore have to consider how the actual transformer deviates from the ideal, and how this affects the circuits in which we employ it. The simplified equivalent circuit of an actual transformer with a 1:1 turns ratio is shown in FIGURE 3(A). Resistors  $R_{\rm p}$  and  $R_{\rm s}$  represent the d.c. resistance of the primary and secondary winding respectively. L<sub>m</sub> is a shunt inductance which is present due to what is called the magnetizing current, i.e. that current which produces flux in the core. L<sub>s</sub> or leakage reactance is due to the fact that the coupling coefficient is less than unity.  $L_s$  is that part of the transformer winding that acts as an inductance. The shunt capacitance Cs represents the stray capacity existing between the windings.

If we consider the low frequency end

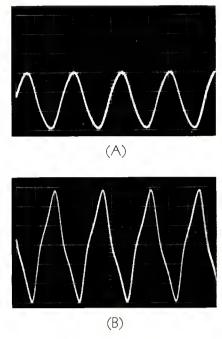


Figure 4. (A) The output waveform below maximum power capability is shown above; (B) the output waveform when the core is in saturation is below.

of the band we can further simplify the circuit by eliminating both Ls whose series reactance is negligible, and  $C_{\rm s}$ whose shunt reactance is very high. If we lump  $R_p$  and  $R_s$  with the generator and load resistances we have a circuit such as in FIGURE 3(B) whose low-frequency roll-off is dependent upon the load and generator resistances assuming a fixed value of shunt inductance. As the generator impedance is increased in the circuit of FIGURE 3(B) the low-frequency roll-off will be more pronounced. The high frequency equivalent circuit is shown in FIGURE 3(C) where the reactance of L<sub>m</sub> is very high and is ignored. In this circuit the high-frequency rolloff is again dependent upon the values of generator and load impedance. When transformers are designed, the values of  $L_m$ ,  $L_s$ , and  $C_s$  are controlled so that the transformer works best in the specified impedance range.

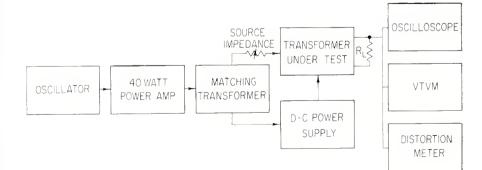


Figure 5. The transformer test set-up used at United Transformer Company.

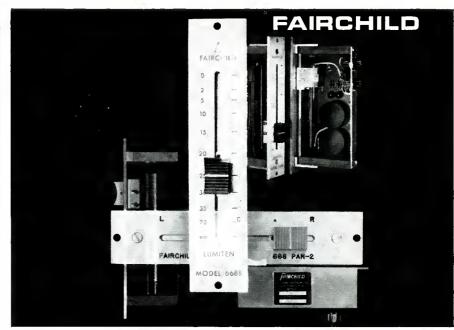


Figure 6. The Test Set-Up of Figure 5 in actual use at UTC.

The effect of the primary and secondary d.-c. resistance is to lower the transformer efficiency. FIGURE 3(D) shows the equivalent circuit as it relates to input/output efficiency in the midband. Part of the power intended for the load is dissipated in resistors R<sub>p</sub> and Rs. Typical losses in a well designed transformer amount to about 1 to  $1\frac{1}{2}$ dB — or we can say that transformer efficiency is about 90 to 85 per cent. UTC design engineers explained to me that it is well within the state-of-the-art to build transformers with higher efficiencies but that size and other economies dictate a compromise design with moderate losses.

An important limitation on transformer operation is the allowable maximum power level. FIGURE 4(A) shows the output waveform of an audio transformer in the mid-band at a level below its maximum capability. When the power level of the transformer is exceeded, the core will saturate and cause distortion. For a given power level the flux density increases proportionally as the frequency decreases, so that core saturation is more of a problem at the low end of the transformer pass band. FIGURE 4(B) shows the output waveform of the same transformer when the core is driven into saturation, and severe waveform distortion is evident.

At UTC, a standard laboratory transformer test set-up is maintained to check the quality of production items and to evaluate new designs. A block diagram of this test set-up, which is used to measure response and distortion, is shown in Figure 5. An oscillator is the signal source, and the 40-watt amplifier provides power gain. The transformer is available to supply the correct voltage range to the transformer and is also used as a d.-c. return path for the power supply which is available for those transformers operating with unbalanced d.c. A variable resistor provides the correct source impedance. Output measuring devices include an oscilloscope, a distortion meter, and vtvm. A photograph of the test set-up in actual use at UTC is shown in FIGURE 6.



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# I heory and Practice

### NORMAN H. CROWHURST

• Sometimes the gaps in theory that "happen" are little bits of information that just never seem to get written up in text or reference books. One of these came to my attention the other day. A radio station engineer asked me about pre-emphasis and de-emphasis in microseconds, which is how response curves are specified for this purpose.

Possibly, as I did, he learned about the response shaping, in terms of dB and phase against frequency, produced by various combinations of R and C elements, and how to calculate this, in terms of the reactance of the capacitor elements. Then, when he came to discussion of pre-emphasis and de-emphasis, for which he may have seen the curves, he finds the response referred to in microseconds.

It's difficult to find anywhere that the relationship between the 3 dB, 45degree phase shift point, by which the response is identified in terms of capacitor reactance, and its designation in

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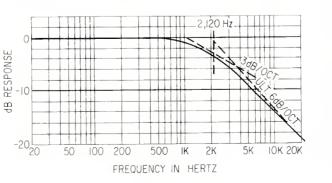
Figure 1. The ideal de-emphasis response, showing its essential features. Not shown is the fact that the loss will be 1 dB at 1,060 Hz, 3 dB at 2,120 Hz (the reference point) and 7 dB at 4,240 Hz.

microseconds is explained.

Let's start with the de-emphasis curve (FIGURE 1). This can be produced by shunting a circuit of known (a.c.) resistance with a capacitor whose reactance has the same value at precisely 2,120 Hz. In its simplest form, this consists of a series resistance and a shunt capacitor. In practice, the effective series resistor is the resistive impedance of the circuit at that point (FIGURE 2).

So far, it's according to the theory we learned. But now we read somewhere that this is a 75-microsecond de-emphasis circuit. Where does that 75microsecond figure come from? It is the time constant of the same R and C combination.

Suppose we are applying the capacitor to the junction between two resistors, each of value 10K (FIGURE 3). The circuit impedance at that point is 5K. So we need a capacitor with a reactance of 5K at 2,120 Hz. By reactance chart, special slide rule, or just applying the



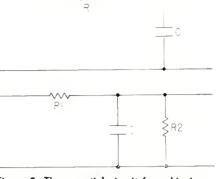


Figure 2. The essential circuit for achieving de-emphasis response. Above, in simple basic form. Below, incorporating input source resistance, R<sub>1</sub>, and output load resistance R2. The equivalent value of R (left) is the parallel combination of R1 and R2.

reactance formula,  $X_c = \frac{1}{2}\pi f C$  (C in Farads, R in Ohms), we can find that the appropriate "C" is 0.015 mFd.

Now, the time constant is R times C. With R in ohms and C in mFd, the time constant is in microseconds. 5,000 times 0.015 figures out to 75 microseconds. In fact any values that will give the 3 dB, 45-degree phase shift point at 2,120 Hz will have a time constant of 75 microseconds. If you do the transposition of formula involved, you'll find that this is obtained by multiplying frequency by 2-pi (6.28) and taking the reciprocal.

Thus 2-pi times 2,120 is 13,330. The reciprocal of this is 0.000,075 seconds, or 75 microseconds.

De-emphasis is easy, pre-emphasis is not quite so easy, but basically it's the same. You put together an R and a C, whose time constant is 75 microseconds, or where the reactance equals the resistance at 2,120 Hz (FIGURE 4) and the current through this combination, for a fixed applied voltage of variable frequency, will have required pre-emphasis characteristic.

But we usually want a voltage output, as well as a voltage input, so we put a resistance at the output end, across which to develop the output voltage (FIGURE 5). And this is where we can complicate matters, by invalidating the result.

As soon as you put in a second resistance, you add another turnover that levels the response (FIGURE 6). The point of this second turnover is found by taking the combined parallel resistance of the terminating resistance and the one the capacitor bypasses.

Thus, if the resistor-capacitor is 5,000 ohms and 0.015 mFd, this sets the lift frequency as 2,120 Hz, or the time constant of 75 microseconds. But if the terminating resistor is 1,000 ohms, the combined parallel value is 833 ohms. A capacitor of 0.015 mFd has this reactance at 12,700 Hz, corresponding to 12.5 microseconds. So the response will level off from this point.

And don't overlook the source resistance. From the viewpoint of response, this is added to the terminating one at the output end. So the response just discussed could result from an output termination of 500 ohms, with a source impedance providing another 500 ohms.

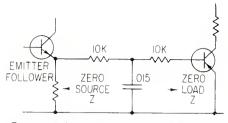
Obviously you cannot push the upper turnover out altogether, because you have to operate the pre-emphasis with finite impedance values. But you need to push it at least beyond the audio range of interest, usually taken as 15 kHz. To yield only 1 dB "loss" at 15 kHz, the second turnover needs to be at 30 kHz, which means the pre-emphasis is obtained with an insertion loss of about 23 dB. If you push the upper turnover to 60 kHz, the loss at 15 kHz will only be 0.25 dB, and the insertion loss is about 29 dB.

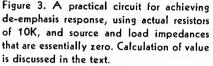
Now maybe all you know all this, but I'll warrant a good proportion of you have been puzzled by it. If that engineer hadn't asked, "What's it mean by rating pre-emphasis or de-emphasis in microseconds?" I wouldn't have thought of writing about it. Too many of us tend to keep quiet about something we imagine we're supposed to know.

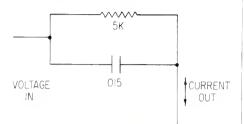
Which reminds me of the time I joined the Audio Standards Committee of the IEEE. They were going over definitions, and it seemed to me they were using "professoreze" to try and impress one another with the changes they thought should be made to bring audio definitions up to date.

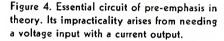
Soon the definition of insertion gain was on the table for discussion. After the old one had been read, I casually commented that I learned that in engineering school, but I still wasn't quite sure what it meant. This comment had a magical effect, for it then appeared that I wasn't alone, and that all of us had acquired a personal interpretation which was slightly different.

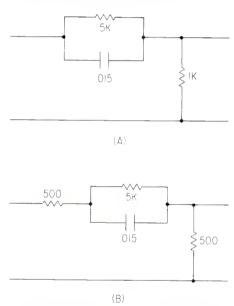
After that the committee really started trying to make the definitions plain and meaningful. But that's another story. As it's come up, though, I'll probably have something to say about the theory and practice of insertion gain in the next issue.

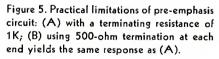












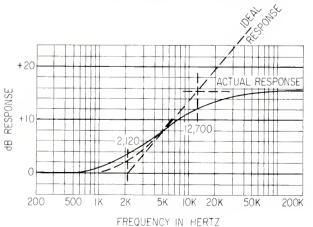


Figure 6. How the practical limitation affects pre-emphasis response. The solidline curve is the response yielded by either circuit of Figure 5.



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# Editorial

OW THAT THE SUMMER HOLIDAY SEASON is in full swing we wonder how many of our readers have noted the deplorable state of publicaddress systems in airports railroad stations, and other transportation terminals. Once you are finally aboard the correct plane or train, there is a tendency to forget the confusion that might well have

existed in getting you there.

We had an occasion to visit the Northwest/Northeast/Braniff Terminal at Kennedy International Airport in New York during the height of the July 4th exodus. Announcements made over the p.a. system were totally unintelligible.

Passengers are apparently a hardy tribe because they managed to find their planes by some mysterious method of visual perception and a primitive form of grapevine communication.

We certainly do not wish to single out this particular installation; it is only an example of what is quite commonplace. By contrast, visitors to European terminals will find sound distribution systems that are exemplary in their ability to get a precise message disseminated.

Vatican Square is an outstanding and world famous example of an outdoor sound system that covers a vast, open expanse with great efficiency, clarity, and effectiveness.

There *are* excellent sound-reinforcement systems to be found in the United States, but they are seldom found in those great public terminals of transportation—where they are so sorely needed.

No one can question the fact that the know-how to create good p.-a. sound exists. Why then is it not prevalent? Is it because sound is considered one place to economize during the construction or modification of buildings?

Those of us engaged in these fields should pressure the architects and other construction specifiers into producing and installing better quality systems. Only when the builders and public officials responsible for terminals are made aware of the ease with which *good* p.a. is attainable, will it become a requirement. R.B.

# The Television Audio Mixer

## MARSHALL KING

A noted television audio mixer tells it like it is in the studio. It's usually past this point that t.v. sound falls to the state it usually is in by the time it gets to home reception.

N THE OUTSET a distinction must be made. I do not refer to the many *filmed* television shows which are done on movie lots (such shows as *Gunsmoke*, *Lassie*, *Bonanza*, *I Love Lucy*, and *I Spy*). I am speaking here of *television* studios where television cameras and television microphones lay down their messages on a two-inch video tape, to produce such shows as *Steve Allen*, *Jackie Gleason*, the defunct *Playhouse 90*, *Joey Bishop*, and *The King Family*.

During the production of a show in a television studio there is an immediacy that is unique to the audio profession. No other audio work that I can think of has a working condition of an exact kind. The old days of live radio drama had it, but there at least audio was the only consideration, and the pressure of immediacy was more than rewarded by the unanimous attention to sound. Not so in television where the picture is king, choreography is queen, and the budget the master of all (with the clock as its spokesman). The immediacy that I speak of does not refer to shows that are done live\*, but to the vast majority of major television programs which are not only taped but which are jackhammered through the time-clock in the interests of expediency, the avoidance of overtime, and the always-hovering knowledge that work on next week's show begins tomorrow. While this puts a hardship on every member of the crew, let's zero in on the audio man, or the mixer.

The pattern at Hollywood Video Center where I work is probably no different than that of any other modern facility. That we can come up with a product each week, or each day, that is acceptable enough for us to allow our names to appear on the credits is, if nothing else, a dubious triumph over frustration, or perhaps a wry tribute to our sense of humor. The best laid plans of audio mixers can, and do, go astray with painful regularity.

Marshall King is audio supervisor and sound mixer at the Hollywood Video Center in Hollywood, California. Some of the shows being done there are The King Family (the author is no relation), Steve Allen, Della Reese, The Talk of Hollywood, the Paul Gregory Specials, Pat Boone, Operation Entertainment, and many commercials for national television. Before joining HVC, Mr. King worked for CBS for seventeen years where he was the mixer on such memorable t.v. shows as Red Skelton, Jack Benny, the Andy Griffith Specials, and Tell It To Groucho. At CBS Radio, before t.v., he worked on Amos n' Andy, Gunsmoke, Gene Autry, and the Lux Radio Theatre.

Why is this so? Why is it that, given an audio mixer who "knows what it should sound like" and who knows his equipment, the final result is often a long way from what *could have been*? Strangely, the reasons are often valid and there is no blame to be cast. When all the elements that lead to this less-than-perfect audio are laid end to end, it can be seen that each one has its rightful place in the scheme of things, and that none is to be impeached out of hand. A conclusion can be reached before we look at details: television audio is a series of compromises.

Presumably, on a television program of any substance, such as a national show that is aired every week, there is a production meeting that takes place a day or two before the show is to be taped. The purpose of such a meeting is for the producers to make their intentions known to the technical crew, and for the technical crew, in turn, to advise the producers as to likelihoods and possibilities. In such an amiable atmosphere where coffee, cigarettes, and attractive assistants prevail, one can only thank his lucky star that he didn't pursue an earlier opportunity for a career in cost accounting.

Really? Let's have a closer look.

In the production meeting the information which comes from the director is top-heavy with enlightenment for the cameramen. Next to this there are pertinent words for lighting, wardrobe, choreography and cue cards. This having been settled, it is apparently time to adjourn and go on to more important activities, for when the director looks at his watch it is a gentle reminder that time is money.

Yet, something is amiss. Nothing has been said about audio. And nothing will be said about audio unless it comes from the mixer. The reason for this is that production people are not, in the main, audio oriented. Why this is so lies beyond the scope of this discussion. My personal hunch is that they, not having come through the ranks of serious radio, do not recognize sound until it is not there. It is something they assume will always be there, much like the reflex action of the breathing apparatus. Only during program silence do they appreciate its value, just as one with emphysema misses his lungs. While this may brand me as a classic case of paranoia, I have too many recollections to bother with argument.

So, what does the mixer do during a production meeting? He listens, anticipates, and speaks. And he speaks the very instant a doubt enters his mind; he does not wait for a pause in the conversation whereby he may interject with discretion. If he sees a situation developing whereby audio will suffer, he challenges at once (ostensibly with a suggestion for a remedy). If his efforts to put forth a workable alternative fall on deaf ears, he must resort to devices most comfortable to himself, such as falling on the floor in a fit, or holding up a card which reads: due to technical difficulties, the audio portion of our program has been interrupted. Please stand by.

Above all, the mixer does not attend a production meeting with the idea of eavesdropping. While it may be easy to not plunge into the melee, it is inevitably disastrous, for later there will be an accounting, though few people think of audio feasibilities at the time of program conception. At best, they assume it will just *be* there, like Mount Everest or the common cold.

I am deliberately belaboring (or possibly exaggerating) the point, for a reason. I am pretending that I am talking to a young man who is going to be a television mixer, and in his understandable exhuberance he thinks his lucky combination of a natural love for music, a taut nerve for drama, and a technical ability in electronics is all he needs to be a sensation in audio. Not so. He must have, in addition to these, the attitude of a close friend of mine who, upon coming home guiltily at four in the morning, shakes his wife out of bed and roars, "You forgot to put the trash out!"

Perhaps this is not an admirable attitude for the true artist, but if producers and directors can ask for economies from the crew, the mixer can be equally expected to protect the quality of his work. And that protection comes from anticipating and speaking up. The blunt truth is that, if the mixer does not make himself known with early dispatch, he will be dismayed to find that many important people are oblivious to audio except during the *post mortem*.

What are some of the obstacles, occurences, or situations which can render the final audio less-than-perfect?

## THE "HIDDEN" MICROPHONE

Normally this is no great problem, for at the present stateof-the-art it is most common for the boom mike to be above the performer's head and just out of the picture. The knowledgeable director will plan his camera shots accordingly, giving the boom time to get in and out as the director cuts from close-up to long shot. Even here it is often a battle of wills, for, the cameraman, in order to better compose his picture, is always fighting for more head-room, while the boom operator tries to help the mixer by keeping the boom mic as low as possible. Why not merely raise the mic and open the fader a little more on the console? Don't they do it that way in the movies? Yes they do, but in the movies the microphone does not hear the movement of four cameras hustling about the stage with the concommitant hiss of the heavy cables they are dragging, or the whir of an iris motor from the nearest camera which may be only inches away, or the shuffle of scenery which is being moved by six stagehands to prepare for the upcoming sequence which will occur in fifty seconds, and it does not hear the bass purr of the air-conditioning ducts which may be as far away as eighty feet or so. All these things are pertinent to live television and have a reason for being. Just to take one: you cannot kill the air conditioning during a "take" in television because inevitably there is an audience sitting there, and if the room becomes the least bit uncomfortable they will either not react favorably to the show (which is death to a performer) or they will

<sup>\*</sup>Other than newscasts and sports events, I know of only one major television show that is done live: Ed Sullivan. When the Sullivan show came to the West Coast for a season and I was assigned as mixer, one of their stage crew told me of the time when Herman's Hermits had the closing spot on the show and, because of a last-minute problem with music, they decided to lip-sync to one of their own records (which had been dubbed to audio tape) rather than sing their intended live number. Apparently Sullivan was not informed of this, for when their number ended and there was still forty seconds to spare, Sullivan called upon the Hermits to sing the last chorus one more time. Since their amplifiers weren't plugged in and there were no microphones in front of them, there followed an awkward period of throat-clearing until Ray Bloch brought his orchestra in for a very long closing theme. Meanwhile, it can be imagined the frantic rewinding of audio tape in the control room as the A-2 man tried to cue to the last forty seconds of the Hermits' song.

merely leave at the first opportunity.

To say another word about the hidden microphone, there is the matter of "lip-sync" and those performers who can't or won't. This refers to those musical numbers whereby, in the ideal situation, the music and voices are pre-recorded on audio tape and played back to the performers during a take, Betty Grable style, during which time they go through the motions of singing. Whether or not they actually sing at this time is irrelevant, since there is not a hot mic around to investigate the issue. The joyous result is perfect sound with no microphone in the picture, even with the camera on a long-shot. There are two reasons why this is not the end-all solution. One is that if ever the performer's lips are the least bit out of sync with his pre-recorded voice, the desired illusion is immediately destroyed, and worse, the viewing audience, from that point hence, will be concentrating on the performer's next slip-up rather than on program content. Another unwelcome surprise, where pre-recorded lip-sync is planned at the outset, is when the performer appears on stage and announces that he or she has absolutely no intention of lip-syncing. After a brief period of jaw-dropping by the producer, director, and talent co-ordinator, there follows a series of accusatory glances which clearly say such things as "But I thought you said," and "Well she told me," and "Her agent assured me just yesterday," and so on. While this may be amusing to those not directly involved, it nearly always spells disaster for audio. For, by this time, the shots have been established, lighting has been set, and all motions have been thoroughly choreographed-at least on the story board if not through rehearsal. To change these things would mean, virtually, to re-write the show. The solution, quite simply, is that she will sing the number live on a boom mic, even though the majority of shots are arranged to show her in the royal ballroom with its beautiful array of chandeliers. To have her sing into a hand mic is unthinkable, since it is a costume number that takes us back to 1743.

Immediately, then, we have a serious audio problem, not only with regard to her voice but with respect to the orchestra as well. For no matter how carefully we have miced the orchestra, the sound will be destroyed once we open the boom to pick up her voice at a distance of twelve feet or more. Nelson Riddle deserves better, and it is for this reason more than any other that audio mixers are known to become busy looking for a dropped coin whenever an arranger passes by.

It is true that many successful pick-ups have been made with a live orchestra, and the singer working on a boom. But here there must be certain conditions which nudge the ideal: the boom can work in tight, the singer has a voice like Bob Goulet, and the music is laid down by a knowledgeable arranger who "writes for the holes". Too often, however, the situation approaches my recurring nightmare whereby I am assigned to a show in which a boom must be used on a wide shot to pick up Eartha Kitt singing Katchaturian's *Sabre Dance* accompanied by the Los Angeles Philharmonic.

Before leaving the subject of the hidden mic, I will say that I have yet to be satisfied with the wireless (transmitter) microphone. The theory of the thing is great, and it looks like they are having good luck with them in the Las Vegas night clubs, but in Las Vegas they do not try to hide the mic under multiple layers of wardrobe, and in television the audiences



Figure 1. The author seated at the audio console of one of Hollywood Video Center's studios. Cartridge and reel-to reel recorders are just out of sight behind him. The glass looks into the studio proper but offers a poor view of the stage. A video monitor above the console shows the picture that is up.

are relatively sober. I eagerly await the acceptable wireless microphone, one that can be interchanged with the conventional cabled mike with no discernment to the listener. Both Vega and Edcor are working on this problem, and I wish them all good luck. In the meantime I have often wondered, quite seriously, why some brilliant young mind doesn't strip away *all* conventional ideas regarding sound pick-up. and come up with something like a sound-modulated laser beam.

# THE PUBLIC-ADDRESS SYSTEM

I have always maintained that the important audience is the seven million people who see and hear our show on television, not the two hundred who come in off the street and sit down in the auditorium. Therefore, rather than cater to the latter by giving them many decibels of p.a. sound, nightclub style, I always instruct the p.a. mixer (who sits in the auditorium with a small console of his own) to feed the very minimum of sound into the house, to make the studio audience stretch their ears. By this I mean to give them no more sound than is necessary to hear the show. The reason for not feeding a hot pa. is, of course, that the performers' mics on stage pick up this secondary source of sound and the audio suffers thereby—not to mention the ever-present threat of feed-back.

One would think that the performer(s) would want nothing more than an assurance that the studio audience is hearing the show clearly and that the miracles of electronics are taking care of the rest, such as recording the show properly on video tape. Not so, for great powers have other ideas. Invariably, the principle performer must have his security blanket, which takes the form of his wanting to *hear himself* on the p.a., no matter where he is on stage. What this does to the recorded sound is obvious. One may ask, "If they don't care, why should you?" The answer is, they *do* care, but if you are flooding the house with p.a. they won't say a word about it *until* they hear one of their shows on the air, at which time they are likely to ask, "What happened with the sound?" And next week it's the same thing all over again-more p.a.

The argument that comes to the mixer from the talent is valid on the surface: if the audience can't hear the material, how can they react to it? This would be a good point except that it makes the assumption that the audience *is not hearing* the material. While you could never convince a performer of this, the mixer is the *first* person to know if the audience is hearing the jokes, for it is always true that, with the p.a. set at a believable low level, the audience *does* laugh at funny jokes and *does not* at unfunny jokes. Further, a loud p.a. does not make dubious material funny. All one has to do to prove the point is to listen to a playback of a comedy show whether the p.a. was set either at moderate or excrutiating makes no difference, and the above premise will be borne out,

These are things we may agree on in private but which. will win us nothing in a confrontation with "talent." Our interest here is: what does the mixer do? Is there a solution? No there is not, for we are dealing here, not with electronics or our ability at the console, but with the psychology of performances. What the mixer does is to handle it much like a chronic case of hives: with great regularity apply your best tonic to keep the thing from getting out of hand. In my case the tonic is to agree amiably when the signal comes to bring up the p.a. but at the same time not to raise it a hair if it will seriously hurt the recorded audio. Obviously, the mixer must know for a fact that the performer is getting a fair deal and that the audience *is* hearing the show. The mixer knows this by the feel of the sound. Too, a good p.a. man will keep the mixer out of trouble, just as will a good boom operator.

One bad practice we try to avoid at Hollywood Video Center is to give the performers a hot p.a. during rehearsals "just because it doesn't hurt anything and it makes them feel good." Therein lies a trap, for what they hear on rehearsal is what they'll expect on the air, and 'tis far better to work out the problem in the afternoon than it is at night when there are other gremlins to cope with.

### AUDIENCE REACTION

This ties in closely with the problems of the p.a., especially on a comedy show, for there are two things which can ruin the audio in such a routine: the p.a. being too hot and the performer not waiting for his laughs. As any radio or television mixer knows, one of your hands is completely tied up with the audience-reaction mics; your hand dare not leave the fader for fear of missing a laugh. At HVC we have eight mics hanging over the audience. These are run into a rack-mounted mixing panel, the output of which comes to the console on a single fader. You can imagine that this fader gets a tremendous workout on six 90-minute Steve Allen shows each week. If the p.a. is too hot and if the performer comes in with another line before the preceding laugh has died down, the result is not good. For, with those eight mics hanging a few feet from the p.a. horns, disaster strikes if another word comes through those horns before the mics are closed again. It is painful sound to the listener, even though he doesn't know exactly what happened. It is also a painful sound to the mixer, and he knows exactly what happened: either he wasn't quick enough, or the performer didn't wait for his laugh.

On a fast-moving show where there is a lot of music integrated with comedy, the mixer will be doing himself a favor by putting the audience-reaction fader on a foot-pedal, operated very much like the huge accelerator on a metropolitan bus. This frees both his hands, a great advantage when the console is split between production dialogue and music. Also, this will give him an out when his critics say that his last show sounded like it was mixed with his feet.

### BUZZES AND HUMS

If there is one thing which the mixer feels is a dirty trick of fate it is when buzzes or hums appear during the show which were not there during the preceding check-out. With hindsight it is easy to see how this can happen, yet there is no guarantee that it won't happen again. If the show is the least bit extemporaneous, and if it is the production policy not to stop the show at any cost (or because of any cost), the mixer merely has to sweat it out when he opens a fader and hears a hum or a buzz. This may have come from an unshielded lamp cord being dragged across his mic lines at the last minute. or it may be due to moisture in a connection if his lines are running across a wet golf course. While he can't stop a golf match, he can stop a studio production, and he must stop it unless the unwarranted annoyance is but the merest peep. He cannot be influenced by past echoes of wails and threats coming from the production booth the last time he stopped the show because of audio trouble. To the bewildered cry, "Why did you kill the audio. . . . it sounded all right to me?" there is only one answer: "It didn't sound all right to me."

Quite naturally, there are times when audio trouble had best be tolerated, as when Jack Benny is in the middle of a monologue and must leave the studio in five minutes to catch a plane; as when an intricate break-a-way set is deliberately being destroyed for the camera and would take hours to set up again; as when Ted Kennedy is delivering a eulogy for his recently departed brother; as when all 37 members of the King Family are lip-syncing San Francisco as we pass under the Golden Gate bridge on a voyage from Honolulu aboard the Lurline. But beyond these extremes, the mixer must take it upon himself to stop the show and correct the audio trouble. And he does so, not by pressing the PL key to the production booth and announcing his intentions (for he will be assured that it doesn't matter and the show must go on), but by swiftly fading down the master gain control so that absolute silence prevails. Then he makes his announcement, and there is naught for anyone to do but allow him to fix the trouble.

# THE QUALITY OF VIDEO TAPE

All the mixer knows is how the sound appears as it leaves his console. While this may sound like a preamble to a cop-out, it is true that the mechanics of a piece of video tape, both emulsion and backing, can be a thorn to both video and audio. It is especially bad for audio, for the sound is relegated to a thin edge of the two-inch-wide tape, the rest being reserved for pictures. And if a particular piece of video tape is going to develop a curl like a Laura Scudder potato chip, where will that curl first take place? On the edges, of course, where the tape is least protected from stresses and strains. Barring emulsion drop-out and windows, audio is inevitably

the first to suffer from a bad piece of tape. Manufacturers guard against this as best they know how, but the machines that grind out the tape are only human and bad stock does occur. The trouble is, no one knows it until the show has been recorded, for no video tape operator monitors the output of his machines during recording. Frankly, it wouldn't do any good anyway, for with the many frequencies of noise and sound that prevail in a video-tape room, one could easily run a Santa Fe train through the premises without attracting too much attention. What often happens is that the sound heard during playback is not at all what the mixer thought he was putting out, and he is obliged to make up the difference during a sweetening session-a poor remedy at best. Herein lies a trap for the unwary mixer, for the tendency is to equalize (or over-equalize) during the recording of a show, on the assumption that past recollections of playbacks have shown that the bass suffers, or the treble is down. No so. If the treble is down today it may not be tomorrow. The rule has to be: don't anticipate a shortcoming. Mix it like it is, and let the other areas of the art come up to that standard. I am sure that every television mixer has been tempted, at some time or other, to boost the bass in an orchestra, knowing that the four-inch speaker in the home receivers will never reproduce what he hears in the booth. This has got to be wrong, for not only are we catering to a deficiency, but what about the one listener in ten thousand who is feeding his tv sound through a good system? Is he to be penalized for his concern? Certainly he is not. Let the manufacturers shoot for a higher mark, for if the audio mixer does not send out good sound it cannot be concocted at the receiving end. This also applies to the quality of video-tape emulsion and backing; if it's not right, let the creator fix it.

# FOUR SPEAKERS AT ONCE?

Everyone gets bored with nostalgia, but I'm going to say it anyway: radio, in its heyday, never had it so good. I mean the kind of radio that brought in the same money that television does today. Jack Benny, Amos n' Andy, Edgar Bergen, Manhattan Merry-Go-Round, First Nighter, The Hit Parade, The College of Musical Knowledge, and Gangbusters. We thought we had problems, but we didn't. We were in a medium that was exclusively audio, and we appealed to the miracle of (not the ears, but) the imagination. What we did not appreciate was that all attention was directed toward sound.

Until 1955 I thought a choreographer was someone who charted the ocean and could sing on key at the same time. My idea of a big problem when I was mixing the *Lux Radio Theatre* (now the *Huntington Hartford*) was how to whip the lag when one performer was reading his lines on our stage and the co-star was reading hers from a stage in New York all of this going on the air live to all cities. We reduced the lag by 50 per cent merely by feeding the audio of both Hollywood and New York to a small booth in Chicago where the show was blended on a two-pot mixer and fed back out to the nation.

But today the mixer is not only one small part of the whole, which is as it should be in television, but he must divide his attention among at least four sources of sound simultaneously. Why four? First of all, he listens to the monitor speaker containing the show audio. Almost as important is the pl (private line) speaker coming from the production booth and containing many surprising messages from the director each minute. One might think that with adequate rehearsal, whereby all moves are laid out in advance, the mixer might disconnect the pl speaker and not listen to it at all during the show. Such stuff are dreams made of, for I can think of scarcely a single show where everything proceeded as planned. Why this is so I cannot say, but it is enough to emphasize that the television mixer had best not disconnect his pipeline, else he will find that everyone else "went thataway."

A third source of information comes to the mixer, continually throughout a show, from his cohorts on stage: the boom operator, the p.a. mixer, the utility man, and the stage managers, not to mention the orchestra leader who is capable of creating new information on the spot. As I say, the mixer can tune these people out, but he will end up doing a show of his own, one having no bearing on the one he is being paid for.

A fourth source of sound which the mixer must adhere to comes from a co-worker in the booth which we call the A-2 man, or second audio. He is another person who, if allowed, can keep the mixer out of trouble. He operates all the playback equipment such as the Ampexes, the cartridges, the turntables, and is, in addition, the recordist. Many bits of information must pass between the mixer and his A-2 man, both for action and verification. A typical example is on the King Family Show, whereby Alice may sing the verse live with piano accompaniment, then go into a lip-sync pre-record on a certain note, with no discernment to the audience. The mixer and the A-2 operator have to work closely on this matter, to say the least. The pre-record tape must be punched up "in tempo" (with no wow) on a fader which was used just previously for a cartridge sound effect, and the musical blend must be homogeneous both as to level and equalization. Beyond this, the pre-recorded tape may have been made in someone else's studios at another time, and the liveness of the two rooms must be matched, falsely of course, by leaving the audience-reaction mics open just enough to liven up the dead tape (not by using the echo chamber).

While every craft has its problems, I can never let it be said by a dilettante that television audio is kept less-thanperfect because the men behind the scenes "want it that way." And for those calm souls who see no problem at all in the abyoe items, I am reminded of a skilled and talented technician in our maintainence shop in years gone by who was absolutely unperturbed by a crisis. He was always proud to say, as he opened his black lunch pail for a leisurely peanut butter sandwich, "I can't imagine a problem that I cannot solve, given enough time." With that he would put away his slide rule and tuck in his bib. I must admit, he was a sharp son-of-a-gun. He knew more about impedance matching and solid-state coupling than I'll ever know, and as part of the ground crew he bordered on genius. But in the cockpit he was no damn good, nor ever will be. The same goes for all critics who are on the outside looking in. Given enough time indeed!

There are no complaints to be made for television audio, only for the limitations imposed by allied factors. But these things will be whipped in time, and many good mixers may become slightly bored if the time comes when everything goes smoothly!

# Television Sound: Panel Discussion

This past spring, a group of nationally known television sound men formed a panel to discuss the questions of television sound. The occasion was a meeting of the New York Chapter of the Audio Engineering Society.

**THE MEMBERS** of the panel included Art Shine of CBS Television. Mr. Shine, who also acted as the meeting's moderator, is in charge of sound mixing on the Ed Sullivan Show.

Mike Shoskis is from Teletape Productions, formerly with CBS Radio where he did the Metropolitan Opera remotes and Newport Jazz festivals. He did the  $\tau v$  audio mixing on ABC's That's Life and has done the Merv Griffin Show and the special Mark Twain Tonight.

Harold Schutzman of CBS Television is formerly an audio operator and now a technical director with a heavy background in news and special events.

Bill Sandreuter is with ABC and is currently doing audio on the Generation Gap and has credits including the last presidential inauguration, and the Apollo and Gemini space shots.

Norman Ogg of NBC has credits that include the Kraft Theater. Circle Theater, Ford specials, and the Robert Montgomery Presents show.

Neil Smith is also with NBC. He has done the Perry Como Show, the Hullabaloo Show, and is currently working on Hidden Faces.

We need to point out that not all the panelists are represented in this transcript partly due to space limitations and partly due to the unsuitability of their largely visual talks in terms of translation onto the printed page.

Art Shine: Let's step back a bit to the beginning of television and remember those Golden Days of Radio. The broadcasting studios were built for the sole purpose of acoustics as in a recording studio; reflection, absorption, and transmission or vibration of sound were and are still of prime importance. In TV, however, audio has always been a final afterthought. Studio space is delegated to scenery, cameras, audience, storage, makeup, dressing rooms, teleprompters, coffee dispensers, and then a small space set aside in the corner of the studio for the orchestra.

Studio floors are highly polished and not only reflect visibly but audibly as well. Scenery and settings are designed solely for their looks, and their synthesized construction usually echoes the "bounce" throughout the studio floor. In radio and recording performers work in fixed and controlled positions, but not so in  $\tau v$ . The microphones and speakers must accommodate the mobility of the medium, and they must follow the performers and beat the traffic on the studio floor.

So, what do we do to overcome these handicaps? Well, one of the things we do in a musical show is to mic the orchestra as tightly as possible, thereby gaining maximum control of all sounds. Then we add reverb to restore or imitate the natural theater quality. We regulate the amount of orchestra leakage to the stage area by means of walls and gobos, and then we feed the pre-mixed orchestra to the vocal area by controlled speakers. We use boom mics of highly directional characteristics to avoid unwanted sounds.



Figure 1. The panel. From the left: Neil Smith, Norman Ogg, Bill Sandreuter, Art Shine, Hal Schutzman, and Mike Shoskis.

You know, the ear is a pretty clever instrument, it screens out the unwanted sounds and only listens to what we want to hear. But the microphone hasn't got that kind of a brain; it's not that clever. It hears all and tells all. So we must overcome that problem with highly directional microphones as best we can. The old radio concept of one, two, or three mics on a band is passe. Because of studio quality and acoustics we don't have that kind of theatrical background to support the one- or two-mic technique. Besides, the contemporary sound of music has so changed that the definition of the orchestra must be more refined, and we must hear, as in recording. The concept of audio pickup is quite different from past years.

"Television sound has one thing going for it. It's the only aspect of television on a technical level that is really artistic enough to be left in the hands of the audio guy. As bad as it is it has that much going for it. It's still up to these guys who have their hands on those controls to mix how much string, and how much bass, how much woodwinds, good or bad."

Those who have been, or know people who have been in recording for radio or  $\tau v$  broadcasting and have left for any length of time and returned, have found it very difficult to catch up to the new trends in audio. Today, for instance, you wouldn't mix an orchestra without micing the drums, balance a band without a tight shot of the flute, or record a vocalist without reverb. Television audio, therefore, must be prepared to accept and accomplish the contemporary sounds.

Today a rock-and-roll group would be nowhere without electronic equipment, from guitars to trumpets, from pianos to drums—all amplified—all of which must be individually miced, equalized, reverbed and finally mixed. So not only have the needs in the studio increased, but so has the equipment in the control room to accommodate these needs.

**Mike Shoskis:** Let's say you've licked your basic pickup problems, and you have a well-balanced orchestra track. Let's see what other obstacles stand between you and the home receiver. Let's take a typical case in which you've recorded an audio tape of an orchestral arrangement that will be played to a live singer, singing on a boom on a floor. The singer invariably wants more level than you are willing to feed to him because of the leakage into the boom mic. That is something that you have to work out with them, or fight out during rehearsals, preferably remaining with the level that you have finally determined.

Now we also find out the importance of the other people who are working with you. The boom operator, if he is right on the action and doesn't swerve, and follows the action carefully, can give you a tremendous hand. The pa man can do the same if he stays with it and he knows the material and what's going to happen. The utility man, whose job very often is to point a loudspeaker at a performer, becomes very important, too. Between these three, they can make the sound for you, if you get a good group of people who know how you work, or know audio themselves, and are not rethreaded video men who were just placed into action because they happened to be there.

"In Chicago, .our audio equipment required. . .a gas mask and a helmet."

# "Stereo sound doesn't match the (tiny) picture."

Now you have your sound on the studio floor. Now, what other obstacles do you have to jump? Let's see what happens to that track that you have fed to the singer and recorded on video tape. In all probability your show will undergo what is known euphemistically as a sweetening session. That's where they put in laughs and applause. This is what makes the show very often sound canned to our ears, because the producer, writer, or director is afraid that the actual live audience reaction is not sufficient. They often will not trust their and the audience's instincts, and feel it necessary to swamp it later with artificial applause and laughs.

So now that is responsible for two more generations of audio, because you make an audio master, and you relay it on to the original video master. You are now up to about fourth generation audio, if you consider that your orchestral background was recorded on quarter-inch tape. Next, it's possible that in laying in commercials another copy has to be made, thus making your original audio tape a fifth-generation copy by the time it reaches the air.

Finally, don't forget that even if you have nursed all the sound all the way down the line, there's a thing known as the compressor at the transmitter. Some stations are notorious for what they do to the sound. It is a heart-rending experience to sit down and listen to a show that you've done, that you know has fairly good audio, and find that the orchestral climaxes are now even with the shuffling of the table and the quiet obscenities of the cameramen muttering to the pl's. It really does happen.

Now, this is not the only thing. You may be doing a field job which, after you have sweated through to produce a reasonable sound, goes down the line in a 5 kHz line. Then you get back in New York and you listen to the air check and have the same slightly sinking feeling.

I'm just mentioning the difficulties of creating the sound. The basic balance is extremely important, of course, because no matter how bad it becomes at the end, the better it is at the starting point the better it will be at the receiver with its proverbial three-inch speaker. Amazingly enough, there *is* some very good  $\tau v$  audio. I listen to some shows and I'm really surprised how that original concept of the audio man does traverse all those obstacles.

**Bill Sandreuter:** Very often when we go out on remotes we find that nobody has given any concern to us, as to where we're going to locate. I've been down to Madison Square Garden and I've had to set my equipment up on the seats, and crowd myself under them, and hope that everything worked out okay. I'd never be able to crawl out and fix anything if anything went wrong.

There's a tremendous lack of communication between production people and the audio department. Very often when we set up a new show we're brought in somewhere downstream. They've already designed the sets and we walk into a studio, and see this great, massive shell, there, with the mc and the group right behind it. The orchestra is always strategically located so that when they really let loose with the brass on a few stings, it will come crashing back to the boom mic.

So you try to point this out to the producer who has spent a fortune for this massive thing out there, and he says, "Geez, it's too late; just do the best you can." This is what we're always faced with. So you go up in the booth and just try. "If you're out of town and hear your show in Miami. . .you won't recognize it, it's really horrible."

Another problem that we always seem to be running into along this line is—"oh, by the way." You're out there, and you're setting up, and the director comes running by, and he says, "Oh, by the way, we've got four zithers and a kazoo, and they're going to be scooting around on motorcycles. The only problem is that they're not going to be here for dress rehearsal."

And he goes on, "You know, we want to hear them." So you go out and you try to make them heard.

There's always the unexpected that you run into. Last summer on the Dick Cavett Show, Dave Garroway came back to New York to me the show. We were told that he wanted to bring his rf mike. This was fine, because I had seen his show many times and his mic is time proven. And I figured we're going to make it tonight with Dave's own little rf mic.

Sure enough, we go to rehearsal, and Dave is out there with his bow tie and he's got the rf mic clipped underneath his shirt, and we go through a beautiful rehearsal. We had no problems whatsoever. Come show time, the hard walls part, the applause is just tremendous, and there stands Dave Garroway with the biggest, fattest, longest, thickest necktie that you ever saw.

Let's see. I've got a couple more points. Audience reaction. Well, particularly on shows like the Generation Gap, the producer always wants to have a lot of audience response I think we touched on canned applause. And we have a new mixer now which the sound effects people at ABC have come up with, which works very well. You only need to put the cartridges in. But we spend a tremendous amount of time balancing up the output of the little cartridge bins trying to match it with the natural sound from the theater.

I prefer to use the natural sound from the theater. When I go into a new show, particularly a game show, or a show like Generation Gap. I try and get together with the person who's going to do the warmup, and have him try to bring the audience along, and tell them that they've got to work too by applauding rapidly. There's nothing more deadly, particularly when you can't see the people, to sit up in the control room and hear a slow clap-clap-clap. So with a little help from the production people (if you cam get it) you can

#### 

get the audience to applaud rapidly. This moves the show along a little faster; that's usually what the producer wants.

Art Shine: Would you gentlemen agree that the so-called five-minute break is probably the best boon to the audio man that was ever invented? It is the time that everybody else seems to take to rest. That's the time to go out and do the work. It's the only available time. If there wasn't a five-minute break I'm sure that we wouldn't get it done.

On the Ed Sullivan Show we do a lot of pre-record, both tapes that we do in the studio and that we bring in from the outside. Of course, one of the helpful aids has been multitrack recorders, We presently deal with only four tracks of audio. Of course we're working in a mono and not a stereo field. company. We discovered that the multi-track machine can be a great aid in this limited rehearsal span, because the multitrack recorder gives us an opportunity to play back and make notes regarding balancing problems.

An orchestra plays one time through and we set the multitrack up so that we can pick up various elements in the orchestra. We can perhaps write up notes as to what exists baritone sax, trombone, etc.—by listening to the various tracks. And we also have an additional opportunity to troubleshoot. For instance, put the vocalist on stage on one track and see how much leakage is there from the orchestra to the stage.

So multi-track is more than just a recording and editing machine; it's also a reference or rehearsal aid.

We've covered a lot of the areas of TV audio. In studios, obviously, there are music shows, dramatic shows, news shows, and educational shows. And in the field come the sporting events and perhaps a church service, concert orchestra at Philharmonic Hall or Carnegie Hall. Under news or special events our network may be called upon to feed the pool. We must therefore, by using one or two microphones, feed hundreds of recorders at any level that they require for a small tape recorder, or a large Ampex, or an air feed. And there are the space missions, riots, or a disaster.

I personally was involved in the recent riots in Grant Park in Chicago during the last Democratic Convention. And we were pretty well gassed, so our audio equipment required not only a microphone and a BN-16, but a gas mask and a helmet. Television audio in the field encompasses a great deal.

The following portion is excerpted from the question period that followed the formal presentations.

**Question:** I have a two-part question. Isn't it true that the AT&T inter-city lines cut off at about 5,000 Hz and therefore, if that is so, isn't TV audio more like AM radio and less like FM radio? And how does this affect your thinking as an audio man?

Art Shine: If it's in Hollywood or New York, they hear a good 15 kHz signal.

Norman Ogg: I've heard a lot of shows and really think New York and Hollywood have got the best sound transmitter-wise because there is no cable to contend with. But if you're out of town and happen to have done the show and hear it in Miami, as an example you won't recognize it, its really horrible.

Question: What amount of sound reinforcement—and could you be specific as to the kind of sound reinforcement—do you provide to the conductor and to the orchestra members, if that's necessary.

Art Shine: The orchestra is quite isolated on the Sullivan Show and therefore it becomes necessary to feed the orchestra somewhere for somebody to hear it. The pa mixer has either multi controls or one fixed control which is the output of the entire orchestra. I give him the output of my mix so that he has to open but one pot. That's for the audience. As far as the orchestra conductor is concerned, he has a head set, and he hears anything I select for him to hear, of by what choice he gives me. There are some orchestra conductors who prefer not to hear their orchestra, who only want to hear the performing talent.

Often an orchestra must hear what goes on on stage to

"Audio men seem to be a strange breed of cat. They certainly don't do it for money, it must be for love."

Prior to air, I often receive tapes from an outside recording

play along with it (the conductor can conduct properly but sometimes it's important that the orchestra get the feel of what's happening onstage). There are two methods. One, of course, is head sets. And the rhythm section does have available to it three, four, five pairs of head sets. Second in the Sullivan orchestra, we feed two speakers to the orchestra, which I control. And I keep it at a low level.

There's another reason why I feed it to them. Being boxed up in that corner they'd go crazy if they didn't hear the comedian on the show, or if they didn't hear Ed Sullivan. And they're kind of nosey about what goes on onstage.

As a matter of fact, in the Sullivan Theater we zig zag sound more than anybody else. When I say zig zag I mean we have a speaker back stage where the performer about to

### "There is a tremendous lack of communication between production people and the audio department."

enter hears the introduction by Ed Sullivan. Because if Sullivan was to say, "Here is, fresh from his trip to so-andso," this performer might want to remark on it when he comes out. So he's got to hear Sullivan say that.

We have a speaker feed, as I said, in the orchestra pit. We have a speaker right on Ed Sullivan's monitor because it's very difficult for Ed Sullivan, twenty, thirty feet away from a performer, to actually hear what that performer is actually singing or saying.

The talent are another problem. The talent feel they can't sing until they hear the sounds coming back. They're used to working in a nightclub where they're surrounded by distorted audio, and they want to hear some of that-it gives them the feeling of singing in the shower, they become a great performer when they hear it. I have had that with Tony Bennett and with Steve and Eydie. They couldn't work without a speaker aimed right at them. So we've developed a few techniques-hidden speakers in the balcony rail—we've even gone so far as to take our regular orchestra speaker on stage. By the way, on stage the performers do get a feed from the orchestra pit, which is a well-balanced mix, the complete mix as I hear it. Controlled by an operator with a vu meter, and he sets the stage level. The vu meter has a stepped control on it. That guy's got to be an audio mixer, or an audio man who's familiar with what he's doing. and he's got to be able to bring the level of that speaker up over the audience reaction when the performer is introduced, and bring that level down when he knows I'm opening the mic to receive the first audio from the performer, till he gets a comfortable level, enough for the performer to hear.

The performer, by the way, that way hears a better sound than he could hear under any circumstances in a theater. If there's an orchestra in the pit he certainly doesn't get a wellbalanced mix. If there's an orchestra off to the side, as we have, and the doors were left open, he wouldn't get a wellbalanced mix. But he does from the speaker.

I don't know of any small speaker that would create the same vivid experience that a performer would get from the experience of an A-7 imitating a live orchestra.

**Bill Sandreuter:** On the Cavett Show and also on Generation Gap we've had some pretty good luck with three column speakers. It's true we have to get them positioned properly, and check out the levels, of course this varies from show to show and week to week. And on Generation Gap we also have a unique problem, the same thing that Art mentioned,



whereby we have to feed the contestants to Jack Barry the mc. We feed the contestants through a little utility speaker that we have under the podium, and then Jack Barry is fed to the contestants so that they can hear the questions, because they're across the stage from one another. We haven't encountered any problems whatsoever. We ride those speakers pretty high.

**Question:** I have a question about perhaps the future of your art. What would be the impact of  $\tau v$  stereo upon  $\tau v$  sound, and do you see any prospects of this in the future?

Art Shine: We at CBS experienced that about two years ago. We made an attempt to broadcast one of the Philharmonic concerts in stereo. For that method we used in New York the WCBS-FM transmitter to carry a multiplex feed while the TV signal was of course a mono mix, so that anybody listening on TV would receive a mono signal.

We had a very strange experience. After the show was over the musicians put in for double pay. They felt they were on the air twice. So the practice was stopped. Our audio console at CBS does not properly handle stereo. I'd love to see it happen as an afterbirth of when everybody has a color receiver and the manufacturers have to sell something else. Sure I'd like to see it happen because it would be something more interesting for us to do.

### "A lot of the talent we're buying today is sorely lacking in mic technique."

Voice: What was the NBC experience with this?

**Neil Smith:** We did the first fifteen minutes of a Perry Como show in mono, feeding only the television network. The next half hour was fed one side of the stereo to the radio-tv network audio, and the other side to the tv audio radio network, and then the last fifteen minutes of the show was back to mono. This was quite a complicated time process, unfortunately they didn't try it again.

There is another thing, too. You've got a small screen. Unless we got a wide video screen to do this on you lose the effect, I think, on television with stereo. Most of the time if you're looking at an orchestra this is primarily what you would be interested in as far as stereo goes. You'd see a small violin player—he's over there in the corner, you know, and out of this speaker comes this big sound and it doesn't match the picture.

# Picture Gallery: West Coast AES Convention

N THIS AND THE FOLLOWING PAGES are the results of our roving camera's visit to the 36th Audio Engineering Society's Convention and Exhibition. It was held at the end of April at the Hollywood-Roosevelt Hotel in Hollywood, California.

The hotel's exhibition space is large, but the AES exhibits managed to fill every available place. Each year's show has grown in proportion to the growth of the audio industry. Continued growth will force the AES to seek expanded space next year in California.

On this page, a montage of the scene at the exhibition. Following, is a display of the *products* shown. Many, of course, premiered at this show.







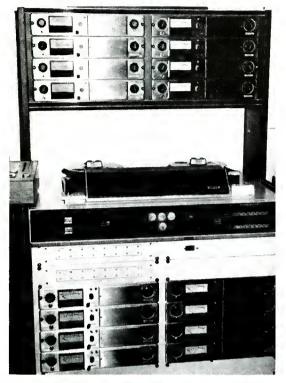








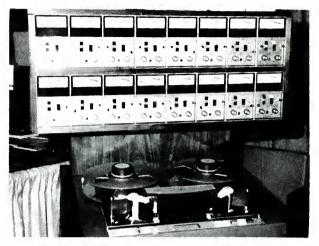
The following illustrations highlight much of the new material shown at the AES Convention. Each product photo is keyed to the Reader Service Card at the back of this issue. Circle the appropriate number for further information to come directly from the manufacturer.



**Ampex.** Eight or sixteen tracks on MM-1000 series machines with their heavy duty and remote overdub transports. Circle 46 on Reader Service Card.



**3M Company.** Remote overdub, sixteen tracks, two-inch tape and the special tape drive system. Circle 33 on Reader Service Card.



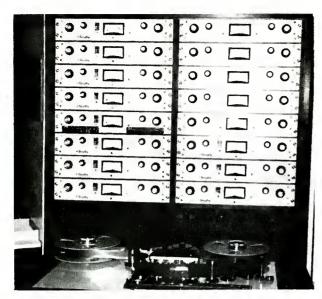
Gauss Electrophysics. Brand-new sixteen-track, two-inch audio recorder. Circle 11 on Reader Service Card.



**AKG.** The C-451 condenser microphone system can be phantom powered from the d.c. supply of a Tandberg 11 portable. Circle 29 on Reader Service Card.



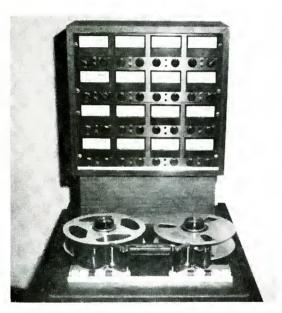
**Harvey Radio.** In one minute up to a two-inch wide tape is automatically entered into and excited from the degaussing field. Circle **44** on Reader Service Card.



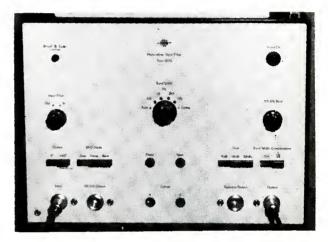
**Scully.** Sixteen tracks on two-inch audio tape with versatile control. Circle 17 on Reader Service Card.



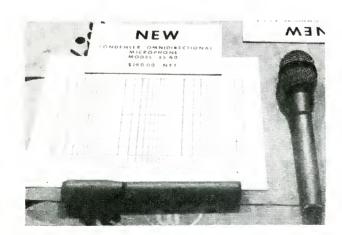
**Stellavox.** The first showing of a French-made professional stereo battery portable. Circle 13 on Reader Service Card.



**Creatronics:** A wholly new sixteen-track two-inch tape recorder for mastering purposes. Circle 36 on Reader Service Card.



**B & K.** The model 2020 heterodyne slave filter may be manually or remotely controlled from 20-20,000 Hz. Circle 21 on Reader Service Card.



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**Electro-Voice.** One of several new condenser microphones from E-V. Circle 45 on Reader Service Card.



**Parasound Orban Associates.** In the middle, a new stereo synthesizer, below a compressor, above a control mixer. Circle 37 on Reader Service Card.



Altec-Lansing. Stepped lights for vu indication and twentyfour tracks in this new console. Circle 20 on Reader Service Card.



**Hewlett-Packard.** Model 204D is the latest low-distortion audio oscillator and model 400F designates a high-sensitivity a.e. vtvm. Circle 34 on Reader Service Card.



**Spectra Sonics.** The Complimiter offers precise high-speed operation with control of slope and release. Circle 31 on Reader Service Card.



**Waveforms.** Combination oscillator and filter set for harmonic distortion measurements. Circle 27 on Reader Service Card.



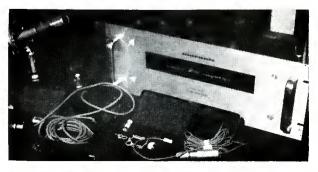
**Shure Bros.** The SM-53 cardioid dynamic, 150 ohms. Circle 12 on Reader Service Card.



**Gately.** Mini-portable consoles and console equipment. Circle 48 on Reader Service Card.



**Gotham Audio.** Total programmed automation is featured in this new Neumann cutting lathe. Circle 32 on Reader Service Card.



**Superscope.** Sony condenser microphones in the front and a Marantz high-power stereo amplifier in back. Circle 40 on Reader Service Card.

# Sound with Images

MARTIN DICKSTEIN

# APOLLO 10-11 COLOR TV FROM SPACE

•Once again, as on the Apollo 9 mission, earth people were treated to a tv show produced from space according to a very precise schedule. Only this time, the programs were transmitted mostly in color.

Westinghouse produced the specially designed color camera and a minimonitor for use on this mission. To indicate the parameters, and some of the stringent requirements involved in the design, let us first look at the black-and-white camera developed for NASA by Westinghouse. (FIGURE 1.)

The environmental conditions set for the operation of the camera were: withstand vibration of 10 to 2,000



Figure 1. This is the type of television camera used by US astronauts transmitting from their craft and the moon. The camera is being held by Stanley Lebar, Westinghouse program manager, it weighs seven pounds, and uses molecular electronic functional blocks in 80 per cent of its circuitry.

db July 1969

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cycles-per-second (or up to 6g) and shocks of more than 8g during launch and landing; pressure levels ranging from sea level to  $10^{-14}$  mm of Hg; temperature variations from 250 degrees F above 0 during the lunar day to =300 degrees F during the lunar night; acoustical noise of 130 dB; resistance to salt spray, 100 per cent oxygen atmosphere; and meteroid bombardment and particle radiation during the time the camera is on the surface of the moon.

The camera had to have facilities for quick-change lenses, scan switching (from 10 f.p.s. to 5/8 f.p.s. for scientific high resolution), be light enough to handle from a restrictive space suit in a small space, and with high enough resolution and extremely wide light range to permit operation during all expected conditions. Tables I and II will indicate the required characteristics and parameters achieved.

In addition to these requirements, consider also that the voltage of the space craft (28 volts) was to furnish the low operating voltages needed and also the -8 kV photo cathode supply. This meant that a special connector had to be developed to perform in a vacuum under the prescribed environmental conditions, special care had to be taken in the insulation of high voltages to prevent corona, and in the shielding of circuits from each other in the small space within the camera package. In extremely reliable method of cooling of the camera of the heat generated within it was required. This necessi-

	Table I. Lens Characteristics		
	Light Level	Lens Field of	Lens
Scene	(Ft lamberts)	View (degrees)	T-Number*
Lunar Surface (Night)	.007 -1.2	30	1.15
Lunar Surface (Day)	20 - 12,600	30	60
Earth & Moon	20 - 12,600	7	60
Spacecraft Interior	.5 - 300	80	5

\*T-Number is the combination of f-number and effect of filtering.

tated design of the package with sufficient surface area and proper surface finish to balance the generated heat and the incoming thermal radiation with the radiated and reflected heat. All fabrication and testing was performed in "clean room" conditions to prevent possible contamination.

The color camera taken on the Apollo 10 mission was scheduled to be used 11 times while the black-andwhite unit was only assigned to one transmission. (This latter pickup was picked up by the NASA facility at Honeysuckle Creek.) All color transmissions were received either by the Madrid or Goldstone receiving antennae and transmitted via Goldstone to NASA's Manned Space Center, Houston, for colorizing of the signal as only the Houston had the necessary equipment to send out a standard broadcast color feed.

The color camera (FIGURE 2) weighed 12 pounds, had a zoom lens, an SEC imaging tube and a rotating color wheel.

The zoom lens with a focal length range of 12.5 mm to 75 mm provided a diagonal field of view variable from 54 degrees to 9 degrees respectively. Aperture stops ranged from f2.2 to f22 and the range of focus was between 20 inches and infinity.

The SEC tube was specially developed by Westinghouse to permit operation of the camera in extremely low light conditions. The standard vidicon is not sensitive enough and the image orthicon is too large and heavy. The SEC tube operates on the principal of secondary electron emission which permits orthicon sensitivity with vidicon operational simplicity and power requirements. (This is the tube scheduled for the Apollo 11 landing.)

Although the color camera is similar to the black-and-white unit in many ways, a most significant difference is in the inclusion of a rotating color wheel in the color unit. Red, green and blue filters are arranged so that the filters pass in front of the imaging tube at a rotational speed of 600 r.p.m. The wheel is divided into six sections to permit two passings of red, blue, and green during each revolution.

The camera then transmits separate red, blue and green images to the ground receiving stations. These images are sent to Houston where special conversion equipment combines the images into a single color picture and produces these at the rate of 30 frames per second which is compatible with the standard rate of commercial television. This *field-sequential* color system is similar to the one pioneered by CBS around 1950. However, the CBS system was not compatible with existing black-and-white transmission used in commercial television. The use of the special conversion equipment at Houston produces com-



Figure 2. Apollo 10 astronauts used such a color camera to take pictures of the moon during their mission. The monitor's screen is about the size of a credit card, with the entire camera weighing less than fifteen pounds.

patible color images.

A further addition to the TV system on the Apollo 10 flight was a miniature monitor (see FIGURE 2). This unit was mounted in the command module (there was no TV transmission at all from the lunar landing module) to assist the astronaut cameramen in determining where the camera was pointed during transmission. The monitor, requiring about 3 watts of power, had a blackand-white viewing screen two inches by two and  $\frac{3}{4}$  inches. The monitor, weighing less than 4 pounds could also be mounted on the camera if desired.

The use of this mini-monitor permits the astronauts to zoom in and focus on the desired scene without help from ground control. To ensure proper settings of the camera variables, the Apollo 10 astronauts were given monitor controls with which they are most familiar and with which the ground evidence can readily identify: Brightness, Contrast, Vertical, and Horizontal.

Barring unforeseen problems with the TV system aboard Apollo 11, the astronauts will be providing 2 hrs., 40 min. of TV from the surface of the moon at 2:12-4:52 a.m. on July 21. The pickup from the surface will be in black and white and will show one astronaut descending to and stepping on the surface of the moon, and erecting an American Flag.

Further surface pickups will show both astronauts gathering lunar rocks and soil samples while also setting out scientific experiments. This transmission will be from the LM or from an umbrella-like high-gain antenna on the surface directly to a 210-foot receiving radio telescope at Parkes, Australia. From Parkes, the signal, converted to standard US television picture, will be

Power Weight Videw Bandwidth Scene Illumination Automatic Control Range Scan Parameters

Aspect Ratio Resolution Operating Temp. Linearity

transmitted to Sydney by microwave and then via satellite to Houston for commercial TV distribution.

The direct feed from the surface of the moon will be one of eight transmissions, the other seven of which will be in color from the command module orbiting at about 70 miles above the moon. The scheduled transmissions will start on July 17 (launch on July 16), July 18, 19, 20, 21, 22, and 23, Only those on July 21 will come during the night-time hours (the color one will be 1:57-2:07 a.m.).

According to the latest information from NASA, the best is yet to come. Plans are now in the making for two three-planet "Grand Tours" in the late

## MANUFACTURERS

## SURPLUS COMPONENTS

Large quantity of level controls, single and dual potentiometers, various fixed resistors, high-wattage power transformers. Also several models of discontinued Acoustech amplifiers (various wattages), f.m. tuner and amplifier kits. Write for complete listing of quantities and prices. Koss Electronics, 2227 North 31 Street, Milwaukee, Wisconsin, Telephone: 414-444-9442. Attn: Bob Bruecker.

Table II. Camera Parameters 6.5 watts, 24 to 32 volts DC input 7.25 lbs 2 Hz to 500 kHz .007 ft-lamberts to 12,000 ft-lamberts 1000-1 10 frames sec. 320 lines frame 5, 8 frames/sec. 1,280 lines/frame 1.3 500 TV lines 0: to plus 130:F 2%

> 1970's. These trips, each taking about 10 years, will consist of the spacecraft flying-by Jupiter, Saturn and Pluto and another flying by Jupiter. Uranus and Neptune. (To illustrate relative distances involved, radio signals from Neptune will take 4 hrs. to reach Earth.) Each of the craft will carry a TV camera for relaying pictures to Earth. As this is the best planet alignment for this purpose in about 180 years, it is a way-out TV special not to be missed.

> We wish to thank Westinghouse, N. Y., and NASA for the technical data. Photographs are courtesy of Westinghouse.

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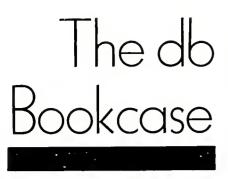
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# People, Places, Happenings



• J. Paul Audet, president of Television & Computor Corp. sets the controls on the latest thing in computerized audio and video broadcast equipment while Carlos V. Girod Jr., secretary and director of r and d looks on. The company, organized late in 1968 has written sales proposals for more than \$3 million worth of broadcast equipment. A data-controlled programmer is the first of a series of products to be marketed by the new company. The firm intends to specialize in the application of computor techniques for the communications and educational markets. It has recently acquired larger quarters in Scotch Plains, New Jersey.



•An announcement from Langevin informs us of the appointment of Jeri Boyd to the position of sales coordinator, executive capacity, In her new position, along with other responsibilities, she will continue the reporting of sales input and products shipped for all audio sales. She has been with the Langevin sales department for the past four years. This new assignment is a planned support to the company's product expansion and industry diversification program. • Record sales and earnings were achieved by **Ampex Corporation** in the fiscal year ended May 3, it was announced today by **William E. Roberts**, president and chief executive officer.

Sales for fiscal year 1969 totaled \$296,319,000, up 27 percent from \$233,-400,000 in fiscal 1968. Pre-tax earnings were \$25,116,000, up 95 percent from \$12,884,000.

Net earnings totaled \$13,702,000, up 79 percent from \$7,665,000. These earnings equaled \$1.35 per share on 10,172,185 average shares outstanding, up 69 percent from 80 cents per share on 9,600,827 shares.

Over the last seven years Ampex has achieved a compounded average annual growth rate in sales of 17 percent; pretax carnings, 20 percent; net earnings, 19 percent; and earnings per share, 18 percent.

• The Institute of Audio Research, Inc., is beginning a new venture in education to provide formal courses in audio recording engineering. The institute was founded in the spring of this year by Albert B. Grundy, a prominent, independent consulting engineer and Irwin Diehl, chief engineer of Caedmon Records, Inc. The institute will offer its first eight week course beginning Sept. 9, 1969, in recording studio theory and practice, with plans to expand course programs by early 1970. Numerous audio recording courses, from elementary practice to seminars in advanced audio engineering are envisioned. Prominent industry figures will be invited to participate as guest lecturers in the advanced seminars.

The technical level of the course beginning in Sept. is oriented toward students currently employed in recording studios and in the early stages of establishing themselves as professionals in the field.

Class meetings will be midtown Manhattan. The Institute's offices are located at 333 Avenue of the Americas, New York City 10014.



• The increased emphasis on professional sound systems by James B. Lansing Sound, Inc. (recently acquired by Jervis Corp.) has resulted in the appointment of L. J. Phillips as assistant manager of JBL's professional products division. In his new position, Mr. Phillips will be developing marketing plans and distribution patters to fill the sound requirements of acoustical consultants, recording studios, sound contractors, and other segments of the professional sound field. To date, the major focus of the 23-year old firm has been directed toward the homeentertainment market, though many IBL systems have found their way into professional installations.

• A recent announcement from **CBS** Labs revealed plans to market audio and video products for the broadcasting industry on an international scale. Negotiations are currently underway with several prospective distributors to make available a complete line of products to European broadcasters. At this publication, marketing and distribution are expected to be underway.

• The 77 year old **Mormon Tabernacle** in Salt Lake City, world famous for its acoustics is installing a custom console audio control system built by **Electrodyne Corporation.** The new installation will handle all audio functions within the 37,500 square-foot structure. This will include recordings of the **Mormon Tabernacle Choir**, as an audio feed for the Choir's network radio and t.v. broadcasts, and as the p.a. system within the Tabernacle. Although Acoustic Research components were designed for home use, they are often chosen for critical professional applications.



Despite decades of experimentation, the manner in which ear and brain process auditory data to sense the direction of a source of sound is still unknown. A new and comprehensive series of experiments now being carried out by researchers at Columbia University may bring us closer to the answer. Under the supervision of Professor Eugene Galanter of the university's Department of Psychology, John Molino and other workers are using elaborate instrumentation to generate precisely controlled signals to synthesize spatial sensations for listeners.

Tests are carried out both indoors and outdoors, necessitating the attachment of wheels to much of the equipment. Part of the apparatus used consists of a "mobilized" AR-3a at lower left in the photograph above, two AR amplifiers (at the bottom of the racks on the table at right), and fifteen mid-range speakers of the type used in the AR-3a. The AR-3a is especially suited to applications of this kind since the uniformity of radiation provides very smooth frequency response on-axis, off-axis, outdoors or in a reverberant room.

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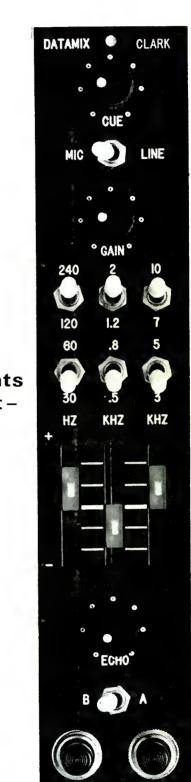
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